WEST Search History

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DATE: Tuesday, April 26, 2005

Hide?	Set Name	Query	Hit Count
		DB=PGPB,USPT; PLUR=YES; OP=ADJ	
☐	L31	(encapsulate or encapsulating) near8 (TICC or BICC)	0
☐	L30	L29 and ITU-T	31
☐	L29	L26 and (media gateway)	90
☐	L28	L26 and l22	0
☐	L27	L26 and NSAP	0
☐	L26	TICC or BICC	430
☐	L25	(encapsulate or encapsulating) near5 address near5 NSAP	0
☐	L24	L22 and TICC	0
☐	L23	L22 and BICC	0
☐	L22	((different or mixed) adj3 address) same (media adj2 gateway)	5
☐	L21	6611584.pn.	1
☐	L20	6785295.pn.	1
☐	L19	6681009.pn.	1
☐	L18	L16 and BICC	0
☐	L17	L16 and BICC and TICC	0
☐	L16	NSAP	177
☐	L15	L14 and TICC	1
☐	L14	L13 and BICC	1
☐	L13	L12 and l11	4
☐	L12	bearer control level	5
☐	L11	call control level	17
		DB=USPT; PLUR=YES; OP=ADJ	
☐	L10	L9 and SS7	2
☐	L9	ITU-T same X.213	2
☐	L8	SS7 addressing	6
☐	L7	open SS7	0
☐	L6	open SS7 addressing	0
☐	L5	open ss7 addressing	0
☐	L4	L3 and SS7	1
☐	L3	L2 or l1	2
☐	L2	5896440.pn.	1

L4: Entry 1 of 1 File: USPT Dec 3, 2002

DOCUMENT-IDENTIFIER: US 6490451 B1
** See image for Certificate of Correction **
TITLE: System and method for providing packet-switched telephony

Detailed Description Text (8):
A managed backbone 205 within CN 240 may be ATM-and/or IP-based, and is utilized to provide packet-switched transport of signaling and bearer data among the nodes within CN 240. On the bearer plane, the backbone network effectively replaces the MSC switching fabrics and inter-MSC trunks of a conventional network of MSCs. On the control plane (referred to alternately as the "signaling plane"), the backbone network replaces signaling transport for the inter-MSC, Signaling System 7 (SS7) signaling of conventional MSC networks. An SS7 Signaling Gateway (SSG) 226 supplies CN 240 with connectivity to SS7 networks, for both ANSI-41 TCAP transactions, and out-of-band call-control signaling (ISUP or CTUP for example). An SS7 network to which SSG 226 connects is typically a separate, circuit-switched network that provides call control and transaction signaling between intelligent network nodes such as CS 214 or WMS 216 within CN 240, and MSC/VLR 228 and HLR 232 within PCN 203.

Detailed Description Text (14):
A Call Server (CS) 214 functions as an MGC for PTMG 225 and WMS 216, and provides wireless- and mobility-independent services, such as call waiting and three-way-call. CS 214 also provides "basic call" functions, such as translations and routing. It should be noted the CS 214 may be derived from an MSC or a Local Exchange (LE), depending on the service set to be offered to mobile subscribers. It is noteworthy that, on the control/signaling plane, WMS 216 and CS 214 together present the appearance of an MSC/VLR to a Public Cellular Network (PCN) 203. While the VLR functionality is entirely provided by WMS 216, MSC functionality is split between WMS 216 and CS 214. WMS 216 presents an MSC interface to the ANSI-41 interface 230 via SSG 226. CS 214 presents MSC-related, SS7 call-control signaling (e.g., ISUP) via SSG 226 and PSTN 210. Media Content Servers 222 include both announcement servers, conferencing servers, and tone generation servers. Such servers may be standalone and/or collocated with WAG 208 and PTMG 225.

Detailed Description Text (19):
As shown in FIG. 3, telecommunication network 300 has signaling interfaces to several external networks including an ANSI-41 signaling network 330, a PSTN 310 and a RAN 302. Each of these is in communication with a core network of functional nodes which collectively manage call setup, mobility, and data transfer functions for a MT within RAN 302. PSTN 310 exchanges call-control signals with a PSTN Trunking Media Gateway (PTMG) 318 utilizing per trunk signaling (R1, R2, C1, etc.). A portion of the SS7 protocol stack within ANSI-41 protocol may be utilized for control signal exchanges between ANSI-41 Signaling Network 330 and SS7 Signaling Gateway (SSG) 326. The remainder of the SS7 protocol stack is utilized to convey call-control signaling between SSG 326 and PSTN 310 (ISUP, CTUP, etc.).

Detailed Description Text (22):
WMS 316 and CS 314 also utilize a media gateway protocol, such as MEGACOP (H.248),

for control message exchanges with media content servers 322. A media gateway control protocol is also employed for control signaling exchanges between WMS 316 and APG 320, WMS 316 and WAG 308, and CS 314 and PTMG 318. ANSI-41 protocol or, an ANSI-41-derived protocol, is utilized for inter-WMS control signaling over either the SS7 network via SSG 326, or the core network. WMS 316 and TP 312 utilize an A-interface-derived protocol, which normalizes, to the greatest extent possible, differences between RF technologies.

Detailed Description Text (62):
CS.sub.H 712 releases pseudo DSO.sub.WMS-H for use in a future call, and translates the call-forwarding DN, resulting in a route to the IP address and UDP port for CS.sub.T 706. As illustrated at step g, CS.sub.H delivers a Setup message to CS.sub.T 706. This Setup message includes the call-forwarding DN, calling number, and the IP address and UDP port associated with termination RTP.sub.PTMG-H. It should be noted that control signaling between CSs 712 and 706 utilizes packet-switched transport within the CN rather than ISUP signaling over the SS7 network, and also specifies the originating RTP termination data instead of a CIC designation for a circuit-switched path. Step h depicts CS.sub.T 706 translating the call-forwarding DN, resulting in the establishment of a route to trunk DSO.sub.PTMG-T which terminates on PTMG.sub.T 704. CS.sub.T /SSG 706 then delivers an ISUP IAM message to a switch within PSTN 702 which terminates the other end of the trunk. This IAM message contains the call-forwarding DN, calling party number, and CIC for DSO.sub.PTMG-T.

Detailed Description Text (84):
Proceeding to step k, CS.sub.H 916 delivers a Setup message which contains the TLDN, calling number, and the IP address and UDP port associated with RTP.sub.PTMG, to CS.sub.S 912. It should be noted that signaling functionality between CSs utilizes packet-switched transport within the CN rather than ISUP signaling over the SS7 network, and also specifies originating RTP termination data instead of a CIC DSO designation. Alternatively, SS7 signaling could be utilized between CSs and associated SSGs, without departing from the spirit or scope of the present invention. At step 1, CS.sub.S 912 translates the TLDN and, finding that it maps to WMS.sub.S 910, routes the call to pseudo trunk DSO.sub.WMS-S. It should be noted that, from the perspective of CS.sub.S 912, WMS.sub.S 910 is a media gateway that terminates pseudo DSO.sub.WMS-S. CS.sub.S 912 delivers a Setup message to WMS.sub.S 910 that contains the TLDN, calling number, pseudo DSO.sub.WMS-S identification, and the IP address and UDP port associated with RTP.sub.PTMG. The RTP.sub.PTMG addressing information is an optimization that obviates an additional MEGACO transaction, as noted with reference to FIG. 8.

Detailed Description Text (105):
The inter-WMS handoff continues at step e, with WMS.sub.A 1108 delivering the equivalent of an IS-41 FacilitiesDirective2 INVOKE message (FACDIR2), directing WMS.sub.T 1112 to initiate a Handoff-Forward task. This message includes identification of MT 1102 (i.e., IMSI/MIN and ESN), billing ID for billing correlation, target cell, and the IP address and UDP port corresponding to APG termination RTP.sub.TO-WAG-T. The IP address and UDP port information is not included within the ANSI-41 standard, but instead replaces the InterMSCCircuitID parameter that would otherwise be utilized in a circuit-switched network context. It should be noted that the FACDIR2 message, as well as other inter-WMS messages may be conveyed over either the CN or the SS7 network. (Recall that WMS functional elements present the appearance of a MSC/VLR to the ANSI-41 SS7 network. Inter-WMS communication over the SS7 network would necessitate that each WMS be provisioned with the point codes [PCs] for the other WMSs).

CLAIMS:

9. The switching control infrastructure of claim 1, wherein said telecommunications environment includes a public cellular network, and wherein said core network

further comprises a Signaling System 7 (SS7) signaling gateway (SSG) providing communicative contact between said WMS and said public cellular network within said control signaling plane.

10. The switching control infrastructure of claim 9, wherein said WMS together with said SSG present the appearance of a MSC/VLR to said public cellular network utilizing TCAP SS7 signaling.

11. The switching control infrastructure of claim 10, wherein said TCAP SS7 signaling consists of either ANSI-41 signaling or GSM Mobile Application Part (MAP) signaling.

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L8: Entry 3 of 6 File: USPT Oct 22, 2002

DOCUMENT-IDENTIFIER: US 6470079 B1
TITLE: System and method for real-time reporting of advertising effectiveness

Detailed Description Text (9):
The protocol used by the AIN is often referred to in the art as "Signaling System 7" or "SS7." SS7 is an addressing protocol that speeds up call processing by transmitting call-connection information out-of-band much faster than traditional call switching. The SS7 protocol has enabled services such as fraud detection, caller ID, store and forward, ring back, concurrent data, etc. The SS7 protocol is well known to those skilled in the art and is described in a specification published by the American National Standards Institute (ANSI).

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L8: Entry 5 of 6 File: USPT Jan 19, 1999

DOCUMENT-IDENTIFIER: US 5862129 A
TITLE: Apparatus and method for the detection and elimination of circular routed SS7 global title translated messages in a telecommunications network

Brief Summary Text (2):
This patent application claims the benefit of earlier field U.S. provisional applications, one titled Apparatus and Method for the Detection and Elimination of Circular Routed Signalling System Number 7 Global Title Translated Messages In a Telecommunications Environment, Ser. No. 60/027913, filed on Oct. 11, 1996; and one titled Apparatus and Method for the Detection and Elimination of Circular Routed Signalling System Number 7 Global Title Translated Messages In a Telecommunications Environment Regardless of the SS7 Addressing Method Utilized, Ser. No. 60/028679, filed on Oct. 18, 1996.

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L32: Entry 3 of 5 File: PGPB Dec 18, 2003

DOCUMENT-IDENTIFIER: US 20030231623 A1
TITLE: Routing system in the next generation open network and method of controlling the routing system

Summary of Invention Paragraph:
[0048] Preferably, the routing server extracts IP address of a terminating media gateway controller by translating the IAM in BICC format received from the originating media gateway controller, transforms the IAM in BICC format into INVITE message corresponding to the SIP-T protocol if the terminating media gateway controller retrieved based on the communication set-up information stored in advance uses SIP-T protocol for communicating with the routing server, and transmits the INVITE message to the terminating media gateway controller.

Detail Description Paragraph:
[0069] If protocol for communication between the routing server 61 and the terminating media gateway controller 33 is SIP-T, the routing server 61 transmits the SIP-T messages to the terminating media gateway controller 33 by transforming all BICC messages received from the originating media gateway controller 31, 34 into SIP-T messages. Further, the routing server 61 transmits the INVITE messages and bearer information to the terminating media gateway controller 33 by transforming the call request signal (IAM) received from the originating media gateway controller 31, 34 into INVITE messages corresponding to SIP-T protocol.

Detail Description Paragraph:
[0070] For instance, if IP address extracted by translating the call request signal is 654.654.654.1, the routing server 61 should transmit the call request signal to another routing server having the above address. Accordingly, the routing server 61 assigns even number as CIC and transmits the call request signal and the bearer information of the originating media gateway 21, 24 to port 3097 of the routing server having the address. If IP address extracted by translating the call request signal is 134.122.52.13 (#33), the routing server 61 transmits transformed information to port 5060 of the terminating media gateway controller 33 by transforming received call request signal (IAM) into INVITE messages corresponding to SIP-T protocol and by transforming all BICC messages into SIP-T messages, because the terminating media gateway controller 33 uses SIP-T protocol.

Detail Description Paragraph:
[0086] In a case where the protocol between the routing server and the terminating media gateway controller 33 is SIP-T protocol, the routing server 61 transforms the IAM in BICC format received from the originating media gateway controller 31, 34 into INVITE messages corresponding to SIP-T protocol and transmits the INVITE messages to the terminating media gateway controller 33.

CLAIMS:

26. The system of claim 24, wherein the routing server is configured to extract IP address of a terminating media gateway controller by translating the IAM in BICC format received from the originating media gateway controller, to transform the IAM in BICC format into INVITE message corresponding to the SIP-T protocol if the terminating media gateway controller retrieved based on the communication set-up

information stored in advance uses SIP-T protocol for communicating with the routing server, and to transmit the INVITE message to the terminating media gateway controller.

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L22: Entry 3 of 5 File: PGPB Mar 25, 2004

DOCUMENT-IDENTIFIER: US 20040057385 A1
TITLE: Methods for discovering network address and port translators

Summary of Invention Paragraph:
[0013] Accordingly, there is provided a method of determining the presence of a Network Address and Port Translator between a Telephony Service Provider Network and a network comprising a Media Gateway, said Telephony Service Provider Network comprising a Media Gateway Controller, said method comprising: instigating a test call between a first endpoint on the Media Gateway Controller acting as a calling party, and an endpoint on the Media Gateway acting as a called party; receiving a reply comprising the local Session Description Protocol of the endpoint of the Media Gateway at the Media Gateway Controller; receiving an empty/noise media packet from the Media Gateway at the first endpoint of the Media Gateway Controller; and comparing the source IP address and port of the empty/noise media packet received at the first endpoint of the Media Gateway Controller with the IP address and port contained in the received local Session Description Protocol of the endpoint of the Media Gateway, wherein if the IP addresses are different or the ports are different then a Network Address and Port Translator is determined to be present, otherwise, if the IP addresses are the same and the ports are the same then a Network Address and Port Translator is determined not to be present.

Summary of Invention Paragraph:
[0014] In a preferred embodiment, if a Network Address and Port Translator is found to present, said method further comprises determining the type of Network Address and Port Translator that is present. Preferably, determining the type of Network Address and Port Translator comprises: changing the remote Session Description Protocol for the call to one with a different IP address, but still on the same Media Gateway Controller, corresponding to a second virtual media endpoint; sending a message to modify the Session Description Protocol to the Media Gateway from the Media Gateway Controller; prompting the Media Gateway to send empty/noise media packets to the changed IP address and port corresponding to the second virtual media endpoint on the Media Gateway Controller; receiving the empty/noise media packets at the second virtual media endpoint on the Media Gateway Controller; comparing the source IP address and port of the empty/noise media packets received at the second virtual media endpoint of the Media Gateway Controller with the source IP address and port of the media packets received at the first virtual media endpoint of the Media Gateway Controller, wherein if the same source IP address and port is used for the media packets sent to the first and the second virtual media endpoints on the Media Gateway Controller, the Network Address and Port Translator is determined to be a Cone Network Address and Port Translator, but if a different source IP address or port is used for the media packets sent to the first and the second virtual media endpoints on the Media Gateway Controller, the Network Address and Port Translator is determined to be a Symmetric Network Address and Port Translator.

Summary of Invention Paragraph:
[0017] According to a fourth aspect there is provided a Media Gateway Controller comprising a first virtual media endpoint, and capable of determining the presence of a Network Address and Port Translator between a Telephony Service Provider Network in which said Media Gateway Controller is located, and a network comprising

a Media Gateway, said Media Gateway Controller operable to: instigate a test call between its first virtual media endpoint acting as a calling party, and an endpoint on the Media Gateway acting as a called party; receive a reply comprising the local Session Description Protocol of the endpoint of the Media Gateway; receive empty/noise media packets from the Media Gateway at its first virtual media endpoint; and compare the source IP address and port of the empty/noise media packets received at its first virtual media endpoint with the IP address and port contained in the received local Session Description Protocol of the endpoint of the Media Gateway, wherein if the IP addresses are different or the ports are different then a Network Address and Port Translator is determined to be present, otherwise, if the IP addresses are the same and the ports are the same then a Network Address and Port Translator is determined not to be present.

Summary of Invention Paragraph:
[0018] In a preferred embodiment if a Network Address and Port Translator is found to be present, said Media Gateway Controller is operable to determine the type of Network Address and Port Translator that is present. Preferably, further comprising a second virtual media endpoint, wherein if a Network Address and Port Translator is found to be present, said Media Gateway Controller is operable to: change the remote Session Description Protocol for the call to one with a different IP address and port, but still on said Media Gateway Controller, corresponding to said second virtual media endpoint; send a message to modify the Session Description Protocol to the Media Gateway; receive empty/noise media data packets at the changed IP address and port, corresponding to its second virtual media endpoint, from the Media Gateway; and compare the source IP address and port of the empty/noise media packets received at its second virtual media endpoint with the IP address and port of the media packets received at its first virtual media endpoint, wherein if the same source IP address and port is used for the media packets sent to said first and said second virtual media endpoints, the Network Address and Port Translator is determined to be a Cone Network Address and Port Translator, but if a different source IP address or port is used for the media packets sent to said first and said second virtual media endpoints, the Network Address and Port Translator is determined to be a Symmetric Network Address and Port Translator.

Detail Description Paragraph:
[0028] The present invention proposes a technique that involves instigating a "test call" during Media Gateway registration, as illustrated in FIG. 1, to determine the existence and type of a NAT. In this test call a virtual media endpoint 3 on the Media Gateway Controller (MGC) acts as the calling party and a real media endpoint 5 on the Media Gateway (MG) acts as the called party. A media endpoint is a source or a sink of media flow, normally Real-time Transport Protocol (RTP) packets. The media endpoints on the MGC are "virtual" since a MGC would not normally have media endpoints and, therefore, they are not "real" endpoints, as such, as in the case of a MG. The MGC is able to deduce the existence of a Network Address Translator (NAT), by examining the media packets (usually RTP packets) sent to the virtual media endpoint 3. If a NAT is determined to be present, the MGC instructs the endpoint 5 to now send media packets to virtual media endpoint 4 on the MGC that has a different IP address than endpoint 3. By comparing the media packets received by endpoint 4 with the packets received previously by endpoint 3, the MGC is able to deduce whether the type of NAT is Cone or Symmetric.

CLAIMS:

1. A method of determining the presence of a Network Address and Port Translator between a Telephony Service Provider Network and a network comprising a Media Gateway, said Telephony Service Provider Network comprising a Media Gateway Controller, said method comprising: instigating a test call between a first endpoint on the Media Gateway Controller acting as a calling party, and an endpoint on the Media Gateway acting as a called party; receiving a reply comprising the local Session Description Protocol of the endpoint of the Media Gateway at the

Media Gateway Controller; receiving an empty/noise media packet from the Media Gateway at the first endpoint of the Media Gateway Controller; and comparing the source IP address and port of the empty/noise media packet received at the first endpoint of the Media Gateway Controller with the IP address and port contained in the received local Session Description Protocol of the endpoint of the Media Gateway, wherein if the IP addresses are different or the ports are different then a Network Address and Port Translator is determined to be present, otherwise, if the IP addresses are the same and the ports are the same then a Network Address and Port Translator is determined not to be present.

10. A method according to claim 9, wherein determining the type of Network Address and Port Translator comprises: changing the remote Session Description Protocol for the call to one with a different IP address, but still on the same Media Gateway Controller, corresponding to a second virtual media endpoint; sending a message to modify the Session Description Protocol to the Media Gateway from the Media Gateway Controller; prompting the Media Gateway to send empty/noise media packets to the changed IP address and port corresponding to the second virtual media endpoint on the Media Gateway Controller; receiving the empty/noise media packets at the second virtual media endpoint on the Media Gateway Controller; comparing the source IP address and port of the empty/noise media packets received at the second virtual media endpoint of the Media Gateway Controller with the source IP address and port of the media packets received at the first virtual media endpoint of the Media Gateway Controller, wherein if the same source IP address and port is used for the media packets sent to the first and the second virtual media endpoints on the Media Gateway Controller, the Network Address and Port Translator is determined to be a Cone Network Address and Port Translator, but if a different source IP address or port is used for the media packets sent to the first and the second virtual media endpoints on the Media Gateway Controller, the Network Address and Port Translator is determined to be a Symmetric Network Address and Port Translator.

23. A Media Gateway Controller comprising a first virtual media endpoint, and capable of determining the presence of a Network Address and Port Translator between a Telephony Service Provider Network in which said Media Gateway Controller is located, and a network comprising a Media Gateway, said Media Gateway Controller operable to: instigate a test call between its first virtual media endpoint acting as a calling party, and an endpoint on the Media Gateway acting as a called party; receive a reply comprising the local Session Description Protocol of the endpoint of the Media Gateway; receive empty/noise media packets from the Media Gateway at its first virtual media endpoint; and compare the source IP address and port of the empty/noise media packets received at its first virtual media endpoint with the IP address and port contained in the received local Session Description Protocol of the endpoint of the Media Gateway, wherein if the IP addresses are different or the ports are different then a Network Address and Port Translator is determined to be present, otherwise, if the IP addresses are the same and the ports are the same then a Network Address and Port Translator is determined not to be present.

25. A Media Gateway Controller according to claim 22, further comprising a second virtual media endpoint, wherein if a Network Address and Port Translator is found to be present, said Media Gateway Controller is operable to: change the remote Session Description Protocol for the call to one with a different IP address and port, but still on said Media Gateway Controller, corresponding to said second virtual media endpoint; send a message to modify the Session Description Protocol to the Media Gateway; receive empty/noise media data packets at the changed IP address and port, corresponding to its second virtual media endpoint, from the Media Gateway; and compare the source IP address and port of the empty/noise media packets received at its second virtual media endpoint with the IP address and port of the media packets received at its first virtual media endpoint, wherein if the same source IP address and port is used for the media packets sent to said first and said second virtual media endpoints, the Network Address and Port Translator is determined to be a Cone Network Address and Port Translator, but if a different

source IP address or port is used for the media packets sent to said first and said second virtual media endpoints, the Network Address and Port Translator is determined to be a Symmetric Network Address and Port Translator.

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L10: Entry 1 of 2 File: USPT Feb 3, 2004

DOCUMENT-IDENTIFIER: US 6687747 B1
TITLE: System and network interoperations using a MIB-based object-oriented signaling protocol

Brief Summary Text (10):
Conventional signaling systems include: Signaling System No. 7 (SS7) which is a signaling protocol widely used to provide message exchange between switches in telecommunication networks; digital subscriber signaling system No. 1 (DSS 1) which is a signaling protocol used in the User-Network Interface (UNI); Resource Reservation Protocol (RSVP) which is an Internet Protocol (IP) network layer signaling protocol used for session control; and session initiation protocol (SIP) which provides for end-to-end control in IP networks. Also, communication between management stations and managed networks in management protocols such as in (CMIP) common management information protocol (see ITU-T recommendation X.711), and simple network management protocol (SNMP) may be considered to be types of signaling protocols.

Brief Summary Text (11):
Typically, most switches in telecommunication networks are signaled in accordance with Common Signaling System No. 7 (SS7). User-network interface (UNI) of ISDN networks supports DSS 1. IP networks use RSVP to support QoS guaranteed multimedia services. The IP telephony and teleconference adopts the H.323 signaling function. Web-based multimedia communication on IP networks uses SIP for ordering and customizing enhanced services on the basis of HTTP services.

Brief Summary Text (12):
One common signaling system used in switches of telecommunication networks is the Common Signaling System No. 7 (SS7). The specifications of the SS7 are published by ITU-T recommendations Q.700-Q.849. The Digital Subscriber Signaling System No. 1 (DSS 1) is also a specification of ITU-T. DSS 1 is offered by the ITU-T recommendation Q.850-Q.999.

Brief Summary Text (17):
For example, SS7 is designed to provide message exchange for interoperations between specific functional entities distributed in network switches. SS7 provides for establishing a signaling channel within a common signaling channel across switches for conveying messages associated with corresponding network control functions. SS7 has been widely used in single-service networks such as telephone networks. However, the bandwidth of the bearer channel is restricted to 64 KB. SS7 does not provide any mechanism for security and access control. Because interoperations between the applications over the SS7 signaling protocol (control functions) have to be designed by message, the amount of messages and control protocols over SS7 will increase exponentially in multimedia multi-service networks. However, it is difficult to implement additional interoperation and interworking between different control functions since the messages are closely related to the corresponding functional components.

Brief Summary Text (58):
An important advantage of the signaling system is that it supports interworking across network layers. A powerful feature of the MIB-based signaling protocol

presented in the invention is the signaling protocol is designed for transparent interworking between different network layers either within or without the same network entity. This is different from current layer-based signaling mechanisms such as SS7 and DSS 1. The signaling protocol of the present invention therefore unifies the interworking across layers and between network entities. On behalf of any network object in any network layer, other network objects in different network entities and different network layers are logically visible and accessible as longer the management information bases are well defined and the operations are authorized.

Other Reference Publication (3):
"Information Technology--Open system Interconnection--Network Service Definition," ITU-T Recommendation X.213 / ISO/IEC 8348 (1995).

Other Reference Publication (14):
Current Signaling System Used in the Switches of Telecommunication Networks in Common Signaling System No. 7 (SS&). The specifications of the SS7 are published by ITU-T Recommendations Q.700-Q.849. (1997).

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L10: Entry 2 of 2 File: USPT Dec 16, 2003

DOCUMENT-IDENTIFIER: US 6665730 B1
TITLE: Method and apparatus for transaction routing in a connection-oriented packet network using a non-fault-tolerant directory server

Brief Summary Text (5):
In communication networks (e.g., telecommunication, packet data, etc), addressing and routing are important network functions enabling efficient communication connectivity. Particularly, for the telecommunication network, addressing and routing are typically performed using Global Title Translation ("GTT") as described in Signaling System 7 ("SS7"), the out-of-band network control signaling system first standardized by the CCITT (now "ITU-T") in 1980. SS7 itself is a connection-less packet network and provides network control signaling for a telecommunications switching network, a circuit-switched network.

Brief Summary Text (6):
Specifically, GTT is a part of the SS7 sub-protocol (i.e., layer), signaling connection control part ("SCCP"). Transaction capabilities application part ("TCAP") uses the signaling connection control part ("SCCP") for the transfer of non-circuit related information between signaling-points in the system, particularly transaction-based information exchange between network entities enabling enhanced services in the telecommunications network. Examples of these services include enhanced dial-1-800 service, automated credit card calling, and virtual private networking. The TCAP protocol enables these services to access remote databases such as service control points ("SCPs") to complete call processing. An SCP can supply the translation and routing information necessary for delivering advanced network services such as translating dialed digits (e.g., 1-800 number) to the required routing number (e.g., routing telephone number). In particular, GTT, performed by signal transfer points ("STP"), comprises the process of translating a global title address of dialed digits to a point code (network code) address and application address (subsystem number) enabling call connectivity within the telecommunications network.

Detailed Description Text (2):
The present invention applies particular routing procedures in a connection-oriented packet network wherein one example applies Global Title Translation ("GTT") techniques. GTT is a part of Signaling System 7 ("SS7") as specified in the 1988 CCITT Recommendations on CCITT, Q.700-Q.795, Geneva 1988. The present invention further relates to ATM network routing particularly ATM addressing functions. These functions are specified in User-Network Interface ("UNI") 3.1, ATM Forum, af-uni-0010.02, 1994; UNI 4.0, ATM Forum, af-sig-0061.000, July 1996; Integrated Local Management Interface ("ILMI") 4.0, ATM Forum, af-ilmi-0065.000, September 1996; ATM Name System ("ANS") 1.0, ATM Forum, af-saa-0069.000, November 1996; Information Technology--Open Systems Interconnection ("OSI") X.213, ITU-T, November 1995; Information Technology--Technology and Information Exchange Between Systems, ISO/IEC 8348, 1993; Private Network-Network Interface, 1.0 Addendum (Soft PVC MIB), ATM Forum, af-pnni-0066.00, September 1996; and Private Network-Network Interface Specification, 1.0, ATM Forum, af-pnni-0055.00, March 1996.

L15: Entry 1 of 1 File: PGPB Feb 13, 2003

DOCUMENT-IDENTIFIER: US 20030031137 A1
TITLE: Signalling in a telecommunications network

Summary of Invention Paragraph:
[0004] ISUP, which deals with the setting-up and control of call connections in a telecommunications network, is closely linked to the E.1/T.1 STM transport mechanisms and does not readily lend itself to use with non-standard transport technologies such as IP and ATM. As such, several standardisation bodies including the ITU-T, ETSI, and ANSI, are currently considering the specification of a signalling protocol for the control of calls, which is independent of the underlying transport mechanism. This is illustrated in FIG. 1 and can be viewed as separating out from the signalling protocol those Bearer Control functions which relate merely to establishing the parameters (including the start and end points) of the "pipe" via which user plane data is transported between nodes, and which are specific to the transport mechanism. The new protocol, referred to as Bearer Independent Call Control (BICC) or Transport Independent Call Control (TICC), retains Call Control functions such as the services invoked for a call between given calling and called parties (e.g. call forwarding), and the overall routing of user plane data. It is noted that signalling traffic at the Call Control level may be sent over a network (IP, ATM, SS7, etc) which is separate from the network over which Bearer Control signalling traffic and user data is sent. However, in some cases a single shared network may be used. As well as TICC, alternative transport independent call control protocols exist including SIP.

Summary of Invention Paragraph:
[0005] The new network architecture resulting from the separation of the Call and Bearer Control levels results in an open interface appearing between a Call Control entity and a Bearer Control entity, where these entities are referred to as a Media Gateway Controller and a Media Gateway respectively. The open interface may be referred to as a Gateway Control Protocol (GCP), examples of which are the MEGACO work of the IETF (MegacoP) and the H.248 work of ITU Study Group 16 (SG16) as well as MGCP. It is envisaged that a given Media Gateway Controller may control several Media Gateways (or indeed a Media Gateway may be controlled by several Media Gateway Controllers).

Summary of Invention Paragraph:
[0013] It will be appreciated that the present invention is applied in the context of a telecommunications network in which the call control and bearer control levels are separate, and where the MG of the bearer control layer is controlled by a Media Gateway Controller (MGC) of the Call Control level.

Summary of Invention Paragraph:
[0017] According to a second aspect of the present invention there is provided a telecommunications system comprising a Media Gateway (MG) at the Bearer Control level and a Media Gateway Controller (MGC) at the Call Control level for controlling the MG,

Brief Description of Drawings Paragraph:
[0021] FIG. 1 illustrates a telecommunications network in which the Call Control

level is independent of the Bearer level;

CLAIMS:

2. A method according to claim 1, wherein said telecommunications network is a network in which the call control and bearer control levels are separate, and where the MG of the bearer control layer is controlled by a Media Gateway Controller, MGC, of the Call Control level.

7. A telecommunications system co prig a Media Gateway, MG, at the Bearer Control level and a Media Gateway Controller, MGC, at the Call Control level for controlling the MG, said MGC comprising means for specifying a Topology Descriptor to define the relationship between a monitoring termination and a monitored termination, the descriptor including a parameter defining whether only information arriving at the monitored termination from outside of the Context, or only information arriving at the monitored termination on from inside of the Context, or both is to be sent to the monitoring termination, and means for sending said Topology Descriptor to the MG to form part of a Context within the MG, and said MG comprising means for receiving said Topology Descriptor and for creating the defined topology within said Context.

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